

Remarks

Applicants have: (a) amended claims 5, 7-12, 14-15 and 17-18 without changing the scope of the claims; and (b) added new claims 19-39. No new matter has been added.

Examiner objected to claim 5. In particular, the Examiner stated:

Claim 5 states “The client apparatus of claim 5.” Claim 5 is objected to because it depends upon itself. Appropriate correction is required.

Applicants have amended claim 5 to depend from claim 4 to correct the inadvertent error pointed out by the Examiner.

In light of the above, Applicants respectfully request the Examiner to withdraw the objection to claim 5.

Examiner rejected claim 5 under 35 U.S.C. § 112. In particular, the Examiner stated:

Claim 5 recites the limitation “the graphical user interface” in line 1. There is insufficient antecedent basis for this limitation in the claim.

Applicants have amended claim 5 to depend from claim 4 to correct the inadvertent error pointed out by the Examiner. As such, Applicants respectfully submit that proper antecedent basis exists for the above-identified limitation.

In light of the above, Applicants respectfully request the Examiner to withdraw this rejection.

Examiner rejected claims 1-14 and 17-18 under 35 U.S.C. 103(a). In particular, the Examiner stated:

Claims 1-14, 17, and 18 are rejected under 35 U.S.C 103(a) as being unpatentable over Katseff et al. (U.S. Pat No. 5,822,537) in view of Kimura (U.S. Patent No. 5,767,863).

As per independent claim 1, Katseff discloses the invention substantially as claimed client apparatus for preparing streaming media received over a non-deterministic delay network for playback or distribution which comprises: a buffer which stores data corresponding to the streaming media

(Figure 1); a rate determiner that determines the time-scale modification playback rate over an interval to control an amount of data in the buffer (col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63); and a user interface which receives a user requested time-scale modification playback rate (col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).

However, Katseff does not explicitly disclose a time-scale modification system that time-scale modifies data output from the buffer at a time-scale modification playback rate.

Kimura discloses a time-scale modification system that time-scale modifies data output from the buffer at a playback rate (Kimura: Figure 3, item 49, col. 6, lines 15-67).

Accordingly, it would have been obvious to one of ordinary skill in the networking art at the time the invention was made to have incorporated Kimura's teachings of the video processing technique with the teachings of Katseff for the purpose of providing improved video processing technique which requires less reduction in bandwidth of the video signal (Kimura: col. 3, lines 44-48).

As per claim 2, Katseff-Kimura further discloses wherein the rate determiner determines the time-scale modification playback rate utilizing the user requested time- scale modification playback rate (Katseff, col. 15, lines 45-65).

As per claim 3, Katseff-Kimura further discloses wherein the user interface further comprises a graphical interface Katseff, col. 13, lines 60-62).

As per claim 4, Katseff-Kimura further discloses wherein the graphical interface further displays one or more of the user requested time-scale modification playback rate, and the time-scale modification playback rate (Katseff: Figure 5, item 550).

As per claim 5, Katseff-Kimura further discloses a client apparatus wherein the graphical interface further displays a range of time-scale modification playback rates which are determined to provide uninterrupted playback (see rejection of claims 3 and 4 above).

As per claim 6, Katseff-Kimura wherein the rate determiner determines the time-scale modification playback rate as a non-linear function of the amount of data (Katseff: col. 2, lines 45-55 and col. 15, lines 55-63, where the teachings of both references disclose balancing the rate at which data is used with the playback rate of audio/video in conjunction with an adaptive control algorithm).

As per claim 7, claim 7 lists all the same elements of claim 1, but in method form rather than apparatus form. Therefore, the supporting rationale of the rejection to claim 1, applies equally as well to claim 7.

As per claim 8, claim 8 is substantially the same as claim 7 and thus rejected using the same rationale. Furthermore, regarding wherein the arrival rate is determined using time-stamps for arriving data (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14., lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).

As per claim 9, claim 9 is substantially the same as claim 8 and thus rejected using the same rationale. Furthermore, regarding wherein the arrival rate is determined by monitoring data arrival times and data packet sizes (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).

As per claim 10, claim 10 is substantially the same as claim 9 and is thus rejected using the same rationale. Furthermore, regarding utilizing time-scale modification to mitigate underflow or overflow in the buffer, or disruption in playback and providing an indication of a current time-scale modification playback rate to the user (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).

As per claim 11, which further comprises steps of: providing an indication of a user requested time-scale modification playback rate (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).

As per claim 12, wherein the step of playing back comprises associating a time-scale modification playback rate with each entry in a playback buffer queue (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).

As per claim 13, wherein the indication comprises a function of recent time-scale modification playback rates (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).

As per claim 14, wherein the step of utilizing comprising ignoring or modifying the user input time-scale modification playback rate when it would interfere with providing continuous playback (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).

As per claim 17, claim 17 is substantially the same as claim 15 and is thus rejected using the same rationale. Furthermore regarding wherein the minimum time-scale modification playback rate is determined as a function of the arrival measure, the consumption measure, an amount of the buffer, and the time interval (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).

As per claim 18, Katseff-Kimura teaches a method for playback of streaming media received over a non-deterministic delay network at a client device which comprises steps of: receiving the streaming media at the client device, which client device includes a CPU; playing back the streaming media; determining a measure of CPU availability; determining a time-scale modification playback rate as a function of the measure of CPU availability; and utilizing time-scale modification to prepare the streaming media for playback (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).

Applicants have amended claims 5, 7-12, 14-15, and 17-18 without changing the scope of the claims. Applicants respectfully traverse this rejection.

Regarding claim 1: Katseff et al. teaches a method for dealing with network congestion or bandwidth limitations that depends on a media encoding technique in which audio and video frames can be dynamically decoupled at broadcast time: see col. 7, lines 1-3; col. 15, lines 1-15; col. 15, lines 30-37; col. 15, line 61 to col. 16, line 10; and col. 16, lines 19-37). In particular, in response to network congestion or bandwidth limitations, Katseff et al. teaches omitting transmission of video (thus eliminating some or all of the video from a presentation), while preserving audio playback at normal speed. Katseff et al. teaches that users prefer audio be unchanged at any expense to video delivery, and that video should be reduced to zero frames per second before audio is modified. However, this is problematic because the result to a user is omission of critical video data. In addition, this is problematic whenever media content is formatted in a manner that does not allow video data to be separated from other media data such as audio by the server.

In addition, the method taught by Katseff et al. is problematic because, when the Katseff et al. device receives no video, the Katseff et al. device can only use audio data to assess network bandwidth. Thus, whenever the Katseff et al. determines, using audio data, that it can request a video frame, the device is uncertain of its arrival time or the impact it will have on transmission of the audio data currently being received.

In contrast to the method taught by Katseff et al., embodiments of the present invention provide a method and apparatus for playing back or presenting of all material in the media work, without dropping video frames or audio frames, by immediately beginning to time-scale modify audio and accompanying video as soon as network congestion is detected. Note that this is counter-intuitive and the opposite of what Katseff et al. teaches.

In addition, and in a nutshell, embodiments of the present invention reduce network congestion due to unsupportable demands for transmission of time-based media content by increasing the total presentation time of that content, rather than decreasing the total amount of media data transmitted in the manner taught by Katseff et al. Advantageously, embodiments of the present invention: (a) do not require a media server to prevent parts of media content from

being transmitted; (b) do not require the media server to alter either the contents or the order of media data that it transmits; and (c) do not require that the media server be able to identify the nature of the media content that it transmits. Further advantageously, in accordance with embodiments of the present invention: (a) the client device or client apparatus may process all of the media content it receives—i.e., it is not necessary to skip or ignore any content; and (b) does not require that the video rendering frame rate—the number of frames of video used to render the video corresponding to a fixed portion of the original multimedia content—be altered (Katseff et al. deletes part of the video—which video is never to be seen again). Further advantageously, embodiments of the present invention: (a) are not limited to media formats in which the media server can identify and separate audio data from other types of media data (for instance video); (b) can be used with formats in which the media server cannot identify or does not separately process any particular media type, such as audio or video; (c) the client device or client apparatus responds to network congestion by immediately modifying the contents of broadcast media, as soon as congestion is detected, rather than only as a last resort; and (d) the client device or client apparatus can process all of the media content broadcast—i.e., it is not necessary to skip or ignore any of that content as taught by Katseff et al.

The different results obtained from the teaching of Katseff et al. and embodiments of the present invention can best be understood by an example of a simple network configuration consisting of the following: a media server, 5 computer workstations which can present media works, and a network which can support a maximum bandwidth of 8 Mb/s. For clarity, we assume the media server contains an audio video work that requires 2 Mb/s of network bandwidth during playback. In this example, the network can support 4 simultaneous viewers (players) presenting the media work from the media server over the network—but not 5. The difference between the teaching of Katseff et al. and embodiments of the present invention becomes apparent when 5 users attempt to view the media work simultaneously. The addition of a 5th user attempting to watch the media work on the 5th workstation means that the 8 Mb/s network is overloaded with requests that total 10 Mb/s. Katseff et al. teaches that the way to deal with such congestion is for the media server to cease sending the video portion of the media work. The method of Katseff et al. has a disadvantage of not reducing the network load until

ceasing to request video data, at which point Katseff et al. no longer has information about the bandwidth of the channel since he is only receiving audio. Thus only the audio data can be examined for assessing network bandwidth. In contrast, embodiments of the present invention counter-intuitively continue with the full presentation of all material without dropping video frames or audio frames by time-scale modifying the audio and accompanying video to create a presentation which is presented in its entirety but at a slower rate—a rate that can be accommodated by the network. Upon the addition of the 5th user using the 5th computer, network congestion arises from the requests for an aggregate of 10 Mb/s (5 * 2Mb/s) to be delivered over a network that can only support 8 Mb/s. The results of such congestion is that the machines become starved for data and this data starvation is noted by the embodiments of the present invention which begin to slow playback on each computer. Thus, although the unmodified presentation is presented in its entirety, it is presented more slowly with greater time between transfers of the portions of the work to be played.

Thus using embodiments of the present invention, a different result is obtained from that obtained using the method of Katseff et al. method; i.e., with embodiments of the present invention, viewers see the entire media work (all media content), and network utilization is maximized without inducing congestion that would otherwise occur on an overloaded network. These results are unexpected in terms of the teaching of Katseff et al.

In addition, as will be set forth in detail below, neither Katseff et al. nor Kimura teach or suggest use of time-scale modification as required by claim 1. As such, even if one were to combine the teachings of Katseff et al. and Kimura in the manner suggested by the Examiner, one would not arrive at the invention of claim 1. As the Examiner agrees, Katseff et al. does not teach or suggest using a time-scale modification system to time-scale modify data output from the audio data buffer at a time-scale modification rate. In addition, Applicants submit that each of the methods Katseff et al. teaches for adjusting the use of audio data is completely different from using a time-scale modification system that time-scale modifies data as required by claim 1. Further, as will discussed in more detail below, Applicants submit that neither Katseff et al. nor

Kimura teach or suggest an apparatus having “a user interface which receives a user requested time-scale modification playback rate” as required by claim 1.

Applicants submit that Katseff et al. does not teach, disclose, hint or suggest a time scale modification system as required by claim 1 (as set forth in the specification, time-scale modification enables digitally recorded audio to be modified so that a perceived articulation rate of spoken passages, i.e., a speaking rate, is modified while maintaining pitch). As set forth in Katseff et al. at col. 2, lines 56-64:

Thus, when network congestion conditions are extreme, the network multimedia system will transmit only audio data without any video data, to the respective workstation. If the audio data does not arrive fast enough over the network to maintain the desired size of the audio buffer when there is no video data being transmitted, the network multimedia system will reduce the speed at which the audio data is played by the workstation until the amount of audio data in the audio buffer has returned to the desired size. (Emphasis added)

As further taught by Katseff et al. at col. 15, lines 17-65:

In a preferred embodiment, the video process utilizes a data buffer monitoring subroutine, illustrated in FIG. 10, to maintain a pre-defined amount of audio and video data in the audio and video buffers 110, 115. . . .

If it is determined during step 1025 that the requested video playback rate has been reduced to the minimum value, network congestion conditions are so extreme that even though no video is being transmitted across the network 20, the audio data is still not arriving fast enough over the network 20 to maintain the desired size of the audio buffer 110. In a preferred embodiment, the data buffer monitoring subroutine will compensate for the delayed arrival of audio data by playing the audio data from the audio buffer 110 at slower than real-time. (Emphasis added)

Katseff et al. states the following regarding playing audio at col. 16, lines 11-37:

The data buffer monitoring subroutine could play the audio at half-speed during step 1030, e.g., by dividing each frames' (sic) worth of buffered audio data into n segments and then playing each segment twice. If it is desired to play the audio at a speed between

half speed and normal speed, not all of the segments n are played twice. Similarly, if it is desired to play the audio at a speed less than half speed, some of the n segments may be played more than twice. ...

In an alternate embodiment, the data buffer monitoring subroutine could play the audio at a reduced speed during step 1030 by utilizing a well-known pitch extraction process, which identifies where the pauses are in the audio and makes the pauses longer.

As the Examiner can readily appreciate from the above, Katseff et al. does not teach, disclose, hint, or suggest using time-scale modification to adjust a rate at which data is extracted from the audio buffer. Further, each of the methods Katseff et al. teaches for adjusting the use of audio data is completely different from time-scale modification.

As explained below, Kimura does not teach using time-scale modification. In particular, Kimura addresses a problem of balancing overall bandwidth capabilities of a hard drive with storage demands of video capture, and Kimura teaches at col. 1, lines 21-23: “an improved technique for transferring video data from a video memory to a separate storage medium, such as a hard disk, in a computer.” Further, Kimura teaches at col. 4, lines 30-36 that “if the average bandwidth of the hard drive is not sufficient to capture all the frames of data being stored in the video memory, a time scaling feature is employed to selectively drop frames of data at periodic intervals to match the average frame capture rate at which frames are being stored in the video memory.” Still further, Kimura teaches at col. 6, lines 25-27: “A time scale control circuit 49 controls the decision/pointer block 46 to drop a field or frame in order to reduce the rate of frames being stored in video memory 48.” Yet still further, Kimura teaches at col. 10, lines 41-51 for a display-only mode:

In step 3, the decision/point circuit 46 is controlled by the time scale control 49 to reduce the frame rate to a predetermined amount, if necessary, to that required (e.g., from 30 frames/second to 27 frames/second) to avoid arbitrarily dropping frames due to throughput limitations of the system. This time scaling may be set by the user or performed automatically by the time scale control 49 detecting unintentional dropping of frames and then incrementally

lowering the frame rate until frames are not unintentionally dropped. This ensures that the displayed image will appear fluid.

Lastly, Kimura teaches at col. 11, lines 9-20:

When the captured frames are played back, the number of dropped frames between the recorded frames may be detected by reading the time stamps, and these captured frames may be repeated as necessary (or interpolated frames may be inserted) to fill in the gap between the captured frames. This places the video image in synchronization with the captured sound track which has also been recorded on the hard disk using conventional techniques along with the video data. The sound track is separated out from the original analog video signal at the front end of the system using a well known demodulator circuit. (Emphasis added)

As the Examiner can readily appreciate from this, Kimura specifically teaches handling audio using conventional techniques. In effect then, Kimura does not teach capturing, storing, or processing audio data using the same, or a similar, method as the one Kimura uses for video, which method for dealing with video is not time-scale modification as required by claim 1).

Thus, as the Examiner can readily appreciate from the above, Kimura does not teach time-scale modification of an audio or an audio-visual work. In fact, all Kimura teaches is reducing video data because incoming video data is arriving too fast for it all to be stored (i.e., there is not enough bandwidth to store the incoming video data in storage before the arrival of new video data).

Thus, the teaching of Kimura regarding audio data is completely different from that of Katseff et al. In particular, while Kimura teaches using conventional methods for storing audio, Katseff et al. teaches a method to increase audio data output from an audio buffer (because incoming audio data is arriving too slowly) wherein copies of the audio data are added to the output to increase the audio data.

In addition, there is no hint or suggestion for combining Katseff et al. and Kimura because Katseff et al. deals with creating more audio data when audio data is arriving too slowly,

and Kimura deals with reducing video data when there is not enough bandwidth to store incoming video data on disk before new video data arrives (i.e., video data is arriving too fast). And, even if one were to combine the teachings of Katseff et al. and Kimura, one would not arrive at the present invention since, among other things; neither of them teaches time-scale modification as required by claim 1.

Lastly, Applicants submit that neither Katseff et al. nor Kimura teach an apparatus having “a user interface which receives a user requested time-scale modification playback rate” as required by claim 1. As taught in Katseff et al. at col. 13, line 61 to col. 14, line 6:

In addition, window 530 includes a playback speed scroll bar 550 that allows a user to control the playback speed, in frames per second, of the recorded presentation, in a known manner. The playback speed scroll bar allows the user to adjust the playback speed from a minimum of zero frames per second, i.e., a still image, up to the maximum recorded frame rate of the video, in either forward or reverse mode. Once the user has selected a video playback speed using the playback speed scroll bar 550, the video process will adjust the rate of data being requested from the storage and retrieval system 70 to the selected playback speed, in addition to making local adjustments to the video and audio outputs of workstation 15.

In light of this, Applicants submit that the video playback speed taught by Katseff et al. is different from the time-scale modification playback rate of claim 1. This is because the time-scale modification playback rate of claim 1 refers to how much to vary the audio articulation rate, and not the video playback rate. Specifically, the term playback speed disclosed in col. 13, line 60 to col. 14, line 6 and FIG. 5 of Katseff et al. refers to a frame-rate of video and the term frame-rate refers to a sampling rate of a live event or recording. For example, a higher frame-rate (or sampling rate) of a video sequence means a greater number of screen updates per unit time, and would therefore be perceived as a higher quality signal. In other words, the term frame-rate refers to a time interval between each frame in a playback. The time-scale modification playback rate of claim 1 is different from the term playback speed of Katseff et al.

because, among other things, the term time-scale modification playback rate of claim 1 is independent of frame-rate. In fact, different playback rates may utilize the same frame-rate.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 2: Claim 2 depends from claim 1. As such, Applicants submit that claim 2 is patentable for the same reasons set forth above with respect to claim 1. In addition, neither Katseff et al. nor Kimura teaches determining a time-scale modification playback rate utilizing a user requested time-scale modification playback rate. In fact, Katseff et al. only teaches determining an audio data output rate (not a time-scale modification playback rate) only on the basis of an amount of data in an audio buffer.

Applicants submit that Katseff et al. merely teaches having a user input frame-rate information for video data. As has been set forth above, this is completely different from a user requested time-scale modification playback rate. Thus, neither Katseff et al. nor Kimura teach or suggest time-scale modification of data at a time-scale modification playback rate that is determined utilizing user requested time-scale modification playback rate information.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 3: Claim 3 depends from claim 2. As such, Applicants submit that claim 2 is patentable for the same reasons set forth above with respect to claims 1 and 2.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 4: Claim 4 depends from claim 3. As such, Applicants submit that claim 4 is patentable for the same reasons set forth above with respect to claims 1-3. In addition, Applicants can find no disclosure or suggestion in Katseff et al. or Kimura for providing a user interface that displays one or more of a user requested time-scale modification playback rate, and a time-scale modification playback rate. As has been discussed above with regard to claim 1, neither Katseff et al. nor Kimura disclose time-scale modification, and Katseff et al. does not provide a graphical display of a user requested time-scale modification playback rate.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 5: Claim 5 depends from claim 4. As such, Applicants submit that claim 5 is patentable for the same reasons set forth above with respect to claims 1-4. Further, neither Katseff et al. nor Kimura teach or suggest in any manner whatsoever a graphical interface that displays a range of time-scale modification playback rates which are determined to provide uninterrupted playback.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 6: Claim 6 depends from claim 1. As such, Applicants submit that claim 6 is patentable for the same reasons set forth above with respect to claim 1. Further, Applicants submit that Katseff et al. does not teach determining a time-scale modification playback rate as a non-linear function of the amount of data. In fact, Katseff et al., as was set forth above, Katseff et al. does not teach determining a time-scale modification playback rate at all. As set forth above, the only thing Katseff et al. teaches is how to reduce the amount of data output from the audio buffer (see col. 15, line 66 to col. 16, line 38).

Applicants submit that Katseff et al. states the following at col. 2, lines 47-52: “In one embodiment, the networked multimedia system compensates for network congestion by using an adaptive control algorithm, which dynamically varies the rate at which video frames are retrieved from the respective file server over the network, in response to network traffic conditions. (Emphasis added)” Applicants submit that the adaptive control algorithm for retrieving video frames from a file server has nothing whatsoever to do with determining a playback rate for time-scale modifying data already in an audio buffer. Instead, the adaptive control algorithm deals with how to retrieve data over the network, and not with playback or distribution of data after it has been received from the network. Further, this teaching has nothing whatsoever to do with determining the playback rate as a non-linear function of the amount of data.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 7: The remarks set forth above with respect to claim 1 apply here as well. In addition, Applicants can find no teaching or suggestion to determine a measure of mismatch between an arrival measure and a consumption measure and utilizing time-scale modification to mitigate effects of the mismatch wherein the arrival measure is determined as a function of an arrival rate of data in a buffer in Katseff et al. or in Kimura. In particular, Katseff et al. only teaches monitoring an amount of data in a buffer, and not its arrival rate. See Katseff et al. at col. 15, lines 19-24 which states: “The data buffer monitoring subroutine will continuously monitor the audio and video buffers 110, 115 during step 1000, until the amount of audio and video data stored in the audio or video buffers 110, 115 drops below a predefined threshold value, as detected by step 1010.” Also see Katseff et al. at col. 15, lines 62-65 which states: “In a preferred embodiment, the data buffer monitoring subroutine will compensate for the delayed arrival of audio data by playing the audio data from the audio buffer 110 at slower than real time.” Also, refer to FIG. 10 of Katseff et al. Hence, Katseff et al. teaches that the “delayed arrival” is determined by monitoring a decrease in data in audio or video buffers. Thus, there is no teaching or suggestion for determining an arrival measure as a function of the arrival rate of data or of using such an arrival measure to determine a measure of mismatch or to use time-scale modification to mitigate effects of the mismatch. Although delays in arrival of data may be the cause of a problem that Katseff et al. attempts to solve, neither Katseff et al. nor Kimura teach utilizing such information, i.e., the arrival rate, in any manner to solve that problem.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 8: The remarks set forth above with respect to claim 7 apply here as well. In addition, the Examiner has asserted that “Furthermore, regarding wherein the arrival rate is determined using time-stamps for arriving data (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14., lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63).” However, Applicants submit that the Examiner is incorrect. Specifically, the only reference to a “time stamp” in Katseff et al. is found at col. 11, lines 51-56, and that reference has nothing to do with using a time stamp to provide a measure of an arrival rate.

Finally, in accordance with Kimura, time stamps are used to insure that video images stay synchronized with unmodified audio (see Kimura at col. 11, lines 13-16 and FIGs. 5 and 8 which show the use of time-stamps to counteract the effect of altering the frame-rate). As a result, Kimura teaches method and apparatus to insure that a "presentation time-line" of a work during playback will be unmodified, and that only a "frame rate" of a video sequence is altered while maintaining the duration of the video sequence. As the Examiner (as well as one of ordinary skill in the art) can readily appreciate from this, Kimura does not teach time-scale modification because time-scale modification, by definition, changes the "presentation time-line" of a work.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 9: The remarks set forth above with respect to claim 7 apply here as well. In addition, the Examiner has asserted that "Furthermore, regarding wherein the arrival rate is determined by monitoring data arrival times and data packet sizes (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63)." However, Applicants submit that the Examiner is incorrect. Specifically, Katseff et al. does not teach monitoring data arrival times or data packet sizes at all.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 10: The remarks set forth above with respect to claim 7 apply here as well. In addition, neither Katseff et al. nor Kimura teach or suggest "determining a time-scale modification playback rate considering one or more of the measure of arrival rate, the measure of a data consumption rate, and user input time-scale modification playback rate requests." In addition to the remarks set forth above with respect to claim 7 and "a measure of arrival rate," Applicants submit that Katseff et al. does not teach or hint determining a measure of a data consumption rate, but only teaches determining an amount of data in a buffer. In further addition, Applicants submit that Katseff et al. does not determine a time-scale modification playback rate for mitigating underflow or overflow in the buffer that entails considering user input time-scale modification playback rate requests since: (a) neither Katseff et al. nor Kimura

teach or suggest using time-scale modification; (b) neither Katseff et al. nor Kimura teach or suggest receiving user input time-scale modification playback rate requests (as set forth above, Katseff et al. merely teaches receiving user input video frame playback rate requests); and (c) neither Katseff et al. nor Kimura teach or suggest using time-scale modification while considering user input time-scale modification playback rate requests. In further addition, Applicants submit that neither Katseff et al. nor Kimura teach or suggest providing an indication of a current time-scale modification playback rate to the user. On the contrary, Katseff et al. only discloses showing a user requested video frame rate. This is completely different from showing a time-scale modification rate provided by the client device in the process of mitigating underflow or overflow in a buffer, or disruption in playback.

In addition, the Examiner has asserted that “Furthermore, regarding utilizing time-scale modification to mitigate underflow or overflow in the buffer, or disruption in playback and providing an indication of a current time-scale modification playback rate to the user (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).” However, Applicants submit that the Examiner is incorrect. Specifically, neither Katseff et al. nor Kimura teach mitigating underflow or overflow in the buffer, or disruption in playback and providing an indication of a current time-scale modification playback rate to the user.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 11: Claim 11 depends from claim 10. As such, Applicants submit that claim 11 is patentable for the same reasons set forth above with respect to claim 10. In addition, as has been set forth above with respect to claim 4, Katseff et al. does not teach providing an indication of a user requested time-scale modification playback rate and only teaches providing an indication of a user requested video frame rate.

In addition, the Examiner has asserted that “As per claim 11, which further comprises steps of: providing an indication of a user requested time-scale modification playback rate (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).” However, Applicants submit that the

Examiner is incorrect. Specifically, neither Katseff et al. nor Kimura teach providing an indication of a user requested time-scale modification playback rate.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 12: Claim 12 depends from claim 10. As such, Applicants submit that claim 12 is patentable for the same reasons set forth above with respect to claim 10. In addition Applicants submit that neither Katseff et al. nor Kimura teach or suggest playing back which comprises “associating a time-scale modification playback rate with each entry in a playback buffer queue” as required by claim 12.

In addition, the Examiner has asserted that “As per claim 12, wherein the step of playing back comprises associating a time-scale modification playback rate with each entry in a playback buffer queue (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).” However, Applicants submit that the Examiner is incorrect. Specifically, neither Katseff et al. nor Kimura teach playing back comprises associating a time-scale modification playback rate with each entry in a playback buffer queue.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 13: Claim 13 depends from claim 10. As such, Applicants submit that claim 13 is patentable for the same reasons set forth above with respect to claim 10. In addition, Applicant submits that neither Katseff et al. nor Kimura teach or suggest “providing an indication of a current time-scale modification playback rate to the user” “wherein the indication comprises a function of recent time-scale modification playback rates” as required by claim 13.

In addition, the Examiner has asserted that “As per claim 13, wherein the indication comprises a function of recent time-scale modification playback rates (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).” However, Applicants submit that the Examiner is incorrect.

Specifically, neither Katseff et al. nor Kimura teach providing an indication which “comprises a function of recent time-scale modification playback rates.”

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 14: Claim 14 depends from claim 10. As such, Applicants submit that claim 14 is patentable for the same reasons set forth above with respect to claim 10. In addition, Applicants submit that neither Katseff et al. nor Kimura teach or suggest utilizing time-scale modification by “ignoring or modifying the user input time-scale modification playback rate when it would interfere with providing continuous playback” as required by claim 14.

In addition, the Examiner has asserted that “As per claim 14, wherein the step of utilizing comprising ignoring or modifying the user input time-scale modification playback rate when it would interfere with providing continuous playback (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63 and Kimura: col. 3, lines 44-48).” However, Applicants submit that the Examiner is incorrect. Specifically, neither Katseff et al. nor Kimura teach utilizing time-scale modification “comprising ignoring or modifying the user input time-scale modification playback rate when it would interfere with providing continuous playback.”

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 17: The remarks set forth above with respect to claim 7 apply here as well. In addition, Applicants submit that neither Katseff et al. nor Kimura teach or suggest a step of utilizing time-scale modification that “comprises determining a minimum time-scale modification playback rate that can be used over a time interval without overflowing a buffer that receives the streaming media; wherein the minimum time-scale modification playback rate is determined as a function of the arrival measure, the consumption measure, an amount of data in the buffer, and the time interval” as required by claim 17. Further, neither Katseff et al. nor Kimura even entertain the concept that there is a minimum time-scale modification playback rate that could be used for the claimed mitigation. In addition, the Examiner asserted:

"Furthermore regarding wherein the minimum time-scale modification playback rate is determined as a function of the arrival measure, the consumption measure, an amount of the buffer, and the time interval (Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63)." Applicant submit that the Examiner is incorrect, and that Katseff et al. does not teach or suggest such a limitation.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Regarding claim 18: The remarks set forth above with respect to claim 7 apply here as well. In addition, Applicants submit that none of the references teach or suggest time-scale modifying streaming data for playback using a playback rate that is a function of a measure of CPU availability. In addition, the examiner asserted that Katseff-Kimura teaches such a method and points to "(Katseff: col. 9, lines 1-22, col. 8, lines 60-67, col. 14, lines 1-54, col. 15, lines 15-67 and col. 16, lines 1-63)." Applicants submit that the Examiner is incorrect since Katseff et al. does not even refer to a CPU in the entire disclosure thereof.

In light of the above, Applicant respectfully requests the Examiner to withdraw this rejection.

Examiner rejected claims 15 and 16 under 35 U.S.C. 103(a). In particular, the Examiner stated:

Claims 15 and 16 are rejected under 35 U.S.C. 103(a) as being unpatentable over Katseff-Kimura and further in view of Allen, U.S. Patent No. 5,652,627.

Regarding independent claim 15, Katseff-Kimura discloses the invention substantially as claimed. Katseff-Kimura further discloses a method for receiving the streaming media at the client device; determining the measure of an arrival rate and a measure of a data consumption rate of the received streaming media; and determining a measure of mismatch between the arrival measure and the consumption measure. However, Katseff-Kimura does not explicitly disclose utilizing time-scale modification to mitigate the effects of the mismatch, wherein the step of utilizing comprises determining a maximum time-scale modification playback rate that can be used over a reporting time interval without draining a buffer that receives the streaming media.

In the same field of endeavor (e.g., system and method for reducing jitter in a packet-based transmission network), Allen discloses utilizing time-scale modification to mitigate the effects of the mismatch.

Accordingly, it would have been obvious to one of ordinary skill in the art at the time the invention was made to have incorporated Alien's teachings of system and method for reducing jitter in a packet-based transmission network with the teachings of Katseff-Kimura, for the purpose of preventing jitters in the delivery of data introduced into a network (Allen: col. 1, lines 66-67 and col. 2, lines 1-7).

As per claim 16, Katseff-Kimura wherein the maximum time-scale modification playback rate is determined as a function of the arrival measure, the consumption measure, an amount of data in the buffer, and the time interval (Allen: col. 1, lines 40-67 and col. 2, lines 1-7).

Applicants have amended claim 15 without changing the scope of the claim. Applicants respectfully traverse this rejection.

Regarding claim 15: Applicants submit that neither Katseff et al. nor Kimura disclose or suggest utilizing time-scale modification to mitigate effects of a mismatch between an arrival rate of data and a departure rate of data from a buffer. In addition, Applicants further submit that Allen does not disclose or suggest utilizing time-scale modification to mitigate effects of a mismatch between an arrival rate of data and a departure rate of data from a buffer.

Applicants submit that Allen notes at col. 1, lines 40-46 that: "A primary concern of MPEG-2 is that the decoder be able to function with a minimum amount of buffer memory. Since prevention of data underflow and overflow in the buffer can only be accomplished by exactly matching the rate of consumption of data from the buffer to the rate of data delivery to the buffer, the decoder clock frequency must exactly match the transmission facility clock frequency." Allen teaches a method for solving this problem in the context of a packet network that is a method for computing a more accurate clock reference and thereby eliminating problems of buffer overflow and buffer underflow that result when there are small frequency drifts between the transmitting device clock and the receiving device clock. Values used to maintain synchronicity between the transmit clock and decode clock are altered in the signal so that accurate presentation rate is maintained and unchanged. As such, Allen provides no teaching or suggestion for utilizing time-scale modification to mitigate effects of mismatch between an arrival measure and a consumption measure. In fact, Allen teaches a method to prevent any alteration in the playback rate from occurring by maintaining synchronous clocks.

In addition, Applicants can find no disclosure or suggestion to utilize time-scale modification in Allen. Specifically, Allen states at col. 4, lines 33-43: “In order to address the inconsistency between the data arrival rate and the data consumption rate, the MPEG-2 standard defines another hypothetical decoder called the video buffering verifier (VBV). … The VBV model allows buffer occupancy to vary between completely full and completely empty and assumes that all data describing a given picture is removed instantaneously from the VBV buffer precisely at the time of the DTS.” As the Examiner can see from this, there is no disclosure at all for utilizing time-scale modification to address the inconsistency between the data arrival rate and the data consumption rate.

The remarks set forth above with respect to claim 7 apply here as well. In addition, Applicants submit that neither Katseff et al. nor Kimura teach or suggest a step of utilizing time-scale modification that “comprises determining a maximum time-scale modification playback rate that can be used over a time interval without draining a buffer that receives the streaming media” as required by claim 15. Further, neither Katseff et al. nor Kimura nor Allen even entertain the concept that there is a maximum time-scale modification playback rate that could be used for the claimed mitigation.

In light of the above, Applicants respectfully request the Examiner to withdraw this rejection.

Regarding claim 16: Claim 16 depends from claim 15. As such, Applicants submit that claim 16 is patentable for the same reasons set forth above with respect to claim 15. In addition, Applicants submit that neither Katseff et al. nor Kimura nor Allen teach or suggest a step of utilizing time-scale modification that comprises determining a maximum time-scale modification playback “wherein the maximum time-scale modification playback rate is determined as a function of the arrival measure, the consumption measure, an amount of data in the buffer, and the time interval.

In light of the above, Applicants respectfully request the Examiner to withdraw this rejection.

Examiner rejected claims 1, 2-4 and 6 on the ground of nonstatutory obviousness-type double patenting. In particular, the examiner stated:

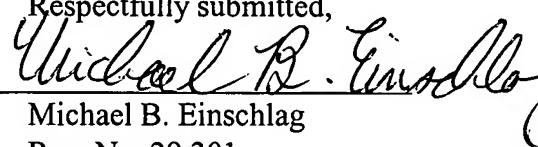
Claims 1, 2-4 and 6 are rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claim 1 of U.S. Patent No. 6,625,655 B2. Although the conflicting claims are not identical, they are not patentably distinct from each other because the differences which include the limitations of claims in the instant application drawn toward a user interface which receives a user requested time-scale modification playback rate, would be obvious to incorporate in order to allow a display for the program playback.

Applicants hereby submit a Terminal Disclaimer.

In light of the above, Applicants respectfully requests the Examiner to withdraw this rejection.

Applicants have added new claims 19-39 to more clearly define the present invention. Applicants respectfully submit that these new claims are patentable over the cited references for at least the reasons set forth above with respect to claims 1-18.

In light of the above, Applicants respectfully submit that all the remaining claims are allowable, and Applicants respectfully request the Examiner to reconsider the case and pass the case to issue. Should the Examiner have any questions or wish to discuss any aspect of the application, a telephone call to the undersigned would be welcome.

Respectfully submitted,
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